Since this response is being filed within two months of the mailing date of the final rejection, the courtesy of an advisory action is respectfully requested.

Claims 1-22 are pending. Claims 1 and 17-22 are rejected under 35 U.S.C. §103 over Borth et al. in view of Hung et al. Claims 2-16 are rejected under 35 U.S.C. §103 over Borth et al. in view of Hung et al. and further in view of Gerson et al. These rejections are respectfully traversed.

The rejections are similar to, and incorporate the same contentions made in the first office action. This response, therefore, incorporates the same rebuttal arguments made in response to the first office action and which have not been addressed by the Examiner in the final rejection.

In particular, the Examiner does not address the salient feature, present in all of the claims, that the present device controls the volume of an audibly reproduced signal ***received from*** a remote transmitter so as to overcome ***background noise surrounding the user*** of the portable telephone. The Examiner may be assuming that what is attenuated at the portable telephone is either the ambient noise or the sound input into a microphone of the portable telephone. Neither is the case.

<u>The rejection of claims 1 and 17-22 under §103 over Borth et al. in view of Hung et al.</u>

The present invention finds its application in, e.g. a portable telephone where it is desired to control the volume of an audibly reproduced signal **received** from a remote transmitter so as to overcome background noise surrounding the user of the portable telephone. Thus, it is an object of the present

invention to control the volume of the reproduced speech signal **sent** from a remote transmitter to the portable telephone to be loud enough to be heard over the ambient noise surrounding the user of the portable telephone. See pages 2-3, page 11, lines 21-23, and page 12, lines 17-21. This feature is clearly set forth in the claims.

Borth et. al. do not teach or suggest both a transmitting and a receiving apparatus nor controlling the volume of ***reproduced sound signals transmitted to a receiver as a function of detected ambient noise surrounding the receiver***. At best, Borth et al. only suggest a noise suppression circuit for "controlling the volume" of the speech entering the microphone of a portable telephone, i.e. by subtracting out noise, as a function of noise detected in the signal received by the microphone of the portable telephone. But Borth et al. do not control the volume of reproduced sound signals being *received* from a remote transmitter. In particular, Borth et al. do not teach or suggest **any** of a vector sum excited linear prediction **(VSELP) encoder** for compressing input speech signals by digital signal processing at a high efficiency, **a transmitting and receiving circuit** for transmitting the compressed speech signals output by the VSELP encoder and for receiving compressed speech signals received from another transmitter and reproducing a corresponding received sound, **noise domain detection means** supplied with analytic patterns produced by the VSELP encoder during compression of the input speech signals for detecting a noise level of a noise domain of the input speech signals, or **means for controlling the sound volume of the reproduced, received sound** responsive to the noise level detected by the noise

domain detection means, all of which are required by claim 1 (and dependent claims 2-16, inclusive). Hung et al.'s VSELP encoder, even if there is some suggestion to make the combination promoted by the Examiner, would still not combine to make a device covered by the claims.

Further, Borth et al. do not teach or suggest a transmitter and a receiver comprising noise level detection means for detecting a sound signal level entering a transmitting microphone as a noise level when there is no transmitting speech input at the transmitter, and control means for **controlling the reproduction volume of a received sound from another transmitter responsive to the noise level** detected by the noise level detection means, as required by claims 17-22, inclusive. Again, Hung et al.'s VSELP encoder, even if combined with Borth et al., would not produce a combination which meets the claims.

<u>The rejection of claims 2-16 under 35 U.S.C. §103 over Borth et al. in view of Hung et al. and further in view of Gerson et al.</u>

The Examiner has again rejected claims 2-16 under 35 U.S.C. §103 as being unpatentable over Borth et al. in view of "Vector Sum Excited Linear Prediction (VSELP) Speech Coding At 8 KBPS", Ira A. Gerson and Mark A. Jasiuk ("Gerson et al."). It is respectfully submitted that Applicants' invention as set forth in claims 2-16 is not obvious in view of the cited prior art.

<u>Claims 2-9</u>

Claims 2 requires that the noise domain detection means employs a first-order linear prediction encoding coefficient as an analytic parameter for each frame of a plurality of frames and

deems a frame to be a noise domain if the first-order linear prediction encoding coefficient is smaller than a pre-set threshold. The Examiner's statement in the first Office Action at page 7, lines 19-24, that "It would have been obvious to a person of ordinary skill in the art at the time the invention was made to use linear prediction encoding coefficients as the analytic parameter to determine whether a noise domain is present because this achieves high speech quality while maintaining reasonable complexity (Pg 1, left col, 4th paragraph)" is a non-sequitur and does not create a prima facie case of obviousness. It is a purely conclusory statement, unsupported by the reference.

Gerson et al. teach speech coding using VSELP parameters, however, neither the Applicants nor the undersigned could find any disclosure in Gerson et al. which teaches or suggests *using a VSELP encoder or any analytic parameters produced by it to detect noise or to control the volume* of a received sound signal. The Applicants respectfully disagree that the portions of Gerson et al. cited by the Examiner support such a contention. There appears to be nothing in those portions of the Gerson et al. text which disclose or suggest detecting noise at all. The Examiner is respectfully requested to clarify which portions of Gerson et al. the Examiner is relying on to support his contention that Gerson et al. teaches or suggests noise domain detection means. All that Gerson et al. appear to disclose is a VSELP encoder and nothing more. The leap to the conclusion that it would be obvious to employ such a VSELP encoder to detect a noise domain is purely in the hindsight of the present specification.

Thus, the rejection of claim 2 (and dependent claims 3-9) must fail on two counts. First, as argued above, Borth et al. do not teach or suggest controlling the volume of a reproduced received sound signal as a function of the ambient noise (as more particularly detailed in the claim language paraphrased above in regard to claim 1) nor do Gerson et al. teach or suggest using a first-order linear prediction encoding coefficient as an analytic parameter for each frame of a plurality of frames and deeming a frame to be a noise domain if the first-order linear prediction encoding coefficient is smaller than a pre-set threshold, as is required by claims 2-9.

Claims 3, 4 and 10, 11

Claims 3 and 10 require that the noise domain detection means employs a pitch gain indicating the intensity of pitch components as the analytic parameter for each frame and deems a frame to be a noise domain if the pitch gain is within a pre-set range. Claim 4, dependent from claim 3, and claim 11, dependent from claim 10, deem a frame to be a noise domain if the pitch gain is zero.

The Examiner cites Gerson et al.'s statements that his VSELP speech decoder uses, as one of three excitation sources, "a long term ('pitch') predictor state" and a "pitch" prefilter as meeting this limitation. The Applicants disagree. There is nothing in Gerson et al. which teaches or suggests determining a noise frame by using pitch gain as an indication of anything, much less as indicating the intensity of pitch components as an analytic parameter for each frame to determine that a frame is a noise domain if the pitch gain is with a pre-set range, or zero, as is required by the claims 3 and 4,

and 10 and 11, respectively. At best, all that Gerson et al. disclose regarding pitch is that a long term filter state (as a pitch predictor) can be used as one of three excitation sources and that the combined excitation signal is processed by an adaptive "pitch" prefilter to enhance periodicity of the excitation signal. See page 4, paragraph VII.

Claims 5 and 12

Claims 5 and 12 require that the noise domain detection means employs a frame power as the analytic parameter for each frame and deems a particular frame to be a noise domain if the frame power for the particular frame is smaller than a pre-set threshold. Again, there is nothing in Gerson et al. which teach or suggest such a feature. Gerson et al. quantizes three excitation gains in two stages. In the first stage, the average speech energy is coded once per frame. This, ultimately, allows the average energy to be factored out so that the three gains can be vector quanitzed efficiently. See page 3, right hand column, first full paragraph. However, there is nothing in this process which suggests a modification of Borth, et al. in view of Hung, et al. to detect a noise domain by employing a frame power as the analytic parameter for each frame and deeming a particular frame to be a noise domain if the frame power for the particular frame is smaller than a pre-set threshold.

Claims 6, 7 and 13, 14

Claims 6, 7 and 13, 14 require that if an amount of change of the frame power between a current frame and a past frame exceeds a pre-set threshold, the noise domain detection means deems the current frame to be a speech domain, even if the current domain is a noise domain. The Examiner finds this suggested by

Gerson et al. at page 4, section IV wherein Gerson et al. describes solving 560 simultaneous equations which result from taking partial derivatives of the total normalized weighted error function with respect to each sample of each basis vector and setting them equal to zero. Because the code frames are not independent, the procedure is iterated in a closed loop fashion. The Examiner apparently contends that solving 560 simultaneous equations in an iterative, closed loop fashion "implies that subtraction is being done between current and past iteration of the frame energies." It is respectfully submitted that this is such a non-sequitur that it is difficult to formulate a response. Iterative solutions of simultaneous equations does not imply determining if an amount of change of the frame power between a current frame and a past frame exceeds a pre-set threshold, and if so, deeming the current frame to be a speech domain, even if the current domain is a noise domain.

Claims 8 and 15, and dependent claims 9 and 16

Claims 8 and 15, and consequently dependent claims 9 and 16, respectively, require a noise level detection means which performs filtering on a noise level output of the noise domain detected by the noise domain detection means. The Examiner points to Gerson et al.'s pitch prefilter in Figure 1 as satisfying this limitation. However, the elements of Gerson et al. prior to the pitch prefilter, i.e. the elements to the left of the pitch prefilter in Figure 1, do not constitute a noise domain *detection* means. They are separate signal sources, not a signal detector. Thus, the pitch prefilter can not serve a function of filtering a noise level of a **detected**

noise domain *output* from a noise domain **detection** means.

For the foregoing reasons it is submitted that all of the claims are now in condition for allowance and the Examiner's early re-examination and reconsideration are respectfully requested.

Respectfully submitted,
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June 17, 1996
Our File: SONY-P4412